Attorney Docket No. 1[illegible]5US01

"Express Mail" mailing label number EL649151594US

Date of Deposit: February 26, 2002

SYSTEM AND METHOD FOR SAP FM DEMODULATION

RELATED APPLICATIONS

[0001] The application herein (docket number 13586US01) references the following applications having docket numbers 13578US01, 13587US01, 13588US01, and 13589US01 filed on the same day as the application herein (docket number 13586US01) was filed.

BACKGROUND OF THE INVENTION

[0002] Certain embodiments of the present invention generally relate to a second audio program (SAP) channel, and in particular, relate to demodulation of a frequency modulation (FM) SAP channel.

[0003] Many television programs are enjoyed by audiences speaking a variety of different languages. Therefore, it is desirable to provide audio content in multiple languages along with video content. Additional audio content is typically easier to encode and transmit than video content, so allowing transmission of multiple audio streams with a single video stream may reduce bandwidth and system complexity requirements.

[0004] Additionally, information may be provided to supplement an audio or audiovisual program. For example, advertisements or promotions related to a television program may be attractive to viewers. Also, customers may enjoy listening to local radio stations through their television sets or may enjoy receiving current weather or traffic reports along with regular television programming.

[0005] SAP channels may be used in audio communications, such as television or radio, for example, to provide a separate audio source in addition to left and right stereo signals or a monoaural audio signal. An SAP channel may be used to duplicate audio content contained on other audio channels or may carry additional information, such as audio content in a second language, descriptive information regarding the audio content, advertising material, local radio stations, weather or traffic reports, frequency-shift keying information, or other additional material unrelated to the main audio content.

[0006] During the 1980s, the Federal Communications Commission (FCC) adopted the format established by the Broadcast Television Standards Committee (BTSC) as a standard for multichannel television sound (MTS). Typically, the BTSC format is used with a composite TV signal that includes a video signal, as well as the BTSC format for sound reproduction.

[0007] The BTSC format is similar to FM stereo, but has the ability to carry two additional audio channels. Left plus right channel mono information is transmitted in a way similar to stereo FM in order to ensure compatibility with monaural television receivers. A 15.734 kHz pilot signal is used, instead of the FM stereo 19 kHz pilot signal, which allows the pilot signal to be phase-locked to the horizontal line frequency. A double sideband-suppressed carrier at twice the frequency of the pilot transmits the left minus right stereo information. The stereo information may be DBX encoded to aid in noise reduction. An SAP channel is located at 5 times the pilot frequency. The SAP channel may be used for second language or independent source program material. A professional audio channel may be added at 6.5 times the plot frequency in order to accommodate additional voice or data.

[0008] Stereo tuners and demodulator units capable of decoding the BTSC format have been on the market for some time. The front end of the units typically includes analog components or integrated circuit chips that cause variation in the amplitude of the composite signal, including the BTSC portion of the signal. This variation in amplitude reduces stereo separation of the right and left channel information carried in the composite signal. Additionally, current stereo tuners and demodulator units attempt to separate an SAP channel using an unnecessarily complicated process with significant hardware cost.

[0009] Further limitations and disadvantages of conventional and traditional approaches will become apparent to one of skill in the art, through comparison of such systems with embodiments of the present invention as set forth in the remainder of the present application with reference to the drawings.

[0010] It is desirable to perform BTSC decoding in the digital domain on a block of an ASIC chip such that the implementation is optimized for reduced complexity and cost. By reducing the complexity, fewer clock cycles are required for processing, and power consumption is also reduced.

[0011] Thus, there is a need for a system that isolates and demodulates an SAP channel without unnecessary noise or variation. Also, there is a need for a simplified and efficient method of demodulating an SAP channel. There is a further need for a system that demodulates an SAP channel of an audio communication without excess hardware. There is a need for a system that utilizes an efficient approximation of the digital FM demodulation equation.

BRIEF SUMMARY OF THE INVENTION

[0012] Certain embodiments of the present invention provides a system and method for SAP FM demodulation. Certain embodiments of the present invention provide a digital FM demodulator that uses a simplified approximation of a digital FM demodulation equation using non-unity delay for simplified demodulation of frequency modulated signals. The system includes a bandpass filter for isolating the SAP signal, a Hilbert filter to produce a copy of the SAP signal phase shifted by 90 degrees, an FM demodulator for demodulating the SAP signal using the phase shifted SAP signal and a delayed SAP signal, and a lowpass filter to eliminate noise from the FM demodulated SAP signal. The system may also include an automatic gain control for normalizing amplitude of FM demodulator input signals. The FM demodulator uses a simplified approximation for easy demodulation of SAP signals. A simplified equation that may be used is $I(n)*Q(n-d)-Q(n)*I(n-d)$, wherein $I(n)$ represents the delayed copy of the SAP information, $Q(n)$ represents the copy of the SAP information with a phase shift, d represents a delay greater than one, and n represents a discrete time index.

[0013] The method includes isolating desired signal information from an audio signal. Then, the method includes phase shifting a copy of the desired signal information from an audio signal and delaying a copy of the desired signal information. The method further includes FM demodulating the desired signal information using the phase shifted copy of the desired signal information and the delayed copy of the desired signal information to produce an FM demodulated signal.

[0014] Certain embodiments also provide a method for simplification of secondary audio program signal demodulation. The method includes using a bandpass filter with a minimal number of coefficients to isolate the secondary audio program signal in a composite audio signal, using a Hilbert filter with a minimal number of coefficients to produce a signal in quadrature phase, and using a simple approximation for FM demodulation of the secondary audio program signal and the signal in quadrature phase.

BRIEF DESCRIPTION OF THE DRAWINGS

[0015] Figure 1 illustrates a SAP demodulation system, formed in accordance with an embodiment of the present invention.

[0016] Figure 2 illustrates a BTSC baseband frequency in accordance with an embodiment of the present invention.

[0017] Figure 3 illustrates a flow diagram for a method of FM demodulating an SAP signal in accordance with an embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

[0018] Figure 1 illustrates a SAP demodulation system 100, formed in accordance with an embodiment of the present invention. The SAP demodulation system 100 includes a BTSC composite audio signal 110, a bandpass filter (BPF) 120, a delay module 130, a Hilbert filter 140, an FM demodulator 150, and a low pass filter (LPF) 160. The output is an FM demodulated signal 170. The system 100 may be used to demodulate digital FM signals, but, for the purposes of illustration, the system 100 will be described in relation to SAP signal demodulation.

[0019] The BTSC composite audio signal 110 feeds into the BPF 120. After passing through the BPF 120, a filtered signal representing the SAP channel travels to both the delay module 130 and the Hilbert filter 140. The delay module 130 produces a signal Xi that is sent to the FM demodulator 150. The Hilbert filter 140 produces a signal Xq that is sent to the FM demodulator 150. A signal from the FM demodulator 150 enters the LPF 160. The FM demodulated signal 170 is generated from the LPF 160.

[0020] In one embodiment, the BTSC composite audio signal 110 is similar to FM stereo but has the ability to carry two additional audio channels. Left plus right (L+R) channel mono information may be transmitted in a way similar to stereo FM in order to ensure compatibility with monaural television receivers. A 15.734 kHz pilot signal may be used, instead of the FM stereo 19 kHz pilot signal, which allows the pilot signal to be phase-locked to the horizontal line frequency. A double sideband-suppressed carrier, at twice the frequency of the pilot, transmits the left minus right (L-R) stereo information. The stereo information may be DBX encoded to aid in noise reduction. An SAP channel may

be located at 5 times the pilot frequency. The SAP channel may be used for second language, advertising, supplement information, or independent source program material, for example. A professional audio channel may be added at 6.5 times the pilot frequency in order to accommodate additional voice or data. Figure 2 illustrates a BTSC baseband frequency in accordance with an embodiment of the present invention.

[0021] The BPF 120 passes a range of frequencies between low and high cut-off frequencies. The BPF 120 may be used to remove stereo and professional audio channels. In one embodiment, the BPF 120 may be a finite impulse response (FIR) filter. A FIR filter generates a response based on an input impulse. The impulse is finite because it does not produce feedback in the filter. The FIR filter receives an input signal and produces a set of coefficients. The input signal and the coefficients form the output signal. Delay may be introduced as the input signal moves past the coefficients. A "tap" of a FIR filter is a coefficient/delay pair. The number of FIR filter taps may relate to the amount of memory in the filter, the scope of the filtering, and the number of calculations involved in the filtering. Thus, fewer taps may indicate fewer calculations and less memory for filtering a signal. The BPF 120 uses a minimal number of taps to isolate the SAP signal. In one embodiment, the BPF 120 is 32-tap FIR bandpass filter. The 32-tap FIR filter reduces the number of calculations involved in filtering the SAP channel from the composite audio signal 110. The 32-tap FIR filter also reduces the hardware and memory used in filtering the audio signal 110 to obtain the SAP channel.

[0022] The Hilbert filter 140 is a filter used to derive a signal in phase quadrature. Signals in phase quadrature have a 90 degree phase difference. The Hilbert filter 140 receives an input signal and produces output with the same frequency response as the input but with a 90 degree phase difference. The Hilbert filter 140 may be a FIR filter or an infinite impulse response (IIR) filter, for example. An IIR filter uses feedback, while an FIR filter does not use feedback. In one embodiment, the Hilbert filter 140 is an 11-tap frequency-domain Remez-designed Hilbert filter.

[0023] The delay module 130 delays the SAP channel signal as a copy of the SAP channel is filtered by the Hilbert filter 140. The delayed signal is used with the phase quadrature signal from the Hilbert filter 140 in the FM demodulator 150. The delay

introduced by the delay module 130 may be a delay d, preferably greater than 1 (non-unity delay).

[0024] The FM demodulator 150 combines the delayed SAP signal from the delay module 130 and the signal in phase quadrature from the Hilbert filter 140 to produce an FM demodulated signal. In one embodiment, the FM demodulator 150 uses a four times (4x) sampling rate. The FM demodulator 150 uses a simplified demodulation equation $m(n)\ \ I(n)*Q(n-d)-Q(n)*I(n-d)$ as an approximation to demodulate the signal, where d is a delay greater than 1. In one embodiment, the delay d is 2, yielding the equation $m(n)\ \ I(n)*Q(n-2)-Q(n)*I(n-2)$. The simplified demodulation equation may also be used for FM demodulation of other digital signals, such as Bluetooth™ digital communication signals and other digital audio signals, for example.

[0025] The LPF 160 passes only low frequencies up to a set cut-off frequency. The LPF 160 is used to remove noise that is out of the desired frequency band from the demodulated signal. In one embodiment, the LPF 160 is a 4^{th} order elliptical LPF.

[0026] In operation, the composite signal 110 is transmitted to the BPF 120. As shown in Figure 2, within the composite signal 110, the SAP signal is centered at five times the pilot signal 25 at a frequency of 78.67 kHz. In order to demodulate this part of the composite audio signal 110, the BPF 120 is applied to the signal 110 to remove the L+R and L-R stereo channels, as well as the professional channel if the professional channel is present in the composite signal 110. Then, the bandpass filtered signal is fed into a Hilbert filter 140 to generate a signal in phase quadrature and is also fed into the delay module 130 to delay the bandpass filtered signal. The delayed signal and signal in phase quadrature are then transmitted to the FM demodulator 150. The FM demodulator 150 applies a simplified demodulation equation to generate a demodulated SAP signal. The demodulated SAP signal is transferred to the LPF 160 to clean up the signal and remove noise outside the desired SAP band. The LPF 160 produces the FM demodulated signal 170.

[0027] In current systems, the FM carrier amplitude at the FM demodulator 150 is not a known constant. In one embodiment, the FM carrier amplitude is adjusted with an automatic gain control (AGC) module (not pictured). In one embodiment, the AGC module includes a comparator and a scaling factor generator. The FM carrier amplitude

is normalized and compared with a programmable reference value. Then, the controlling scaling factor either increments or decrements the FM carrier signal from the normalized value based on the comparator results which are either larger than or less than the reference value.

[0028] Figure 3 illustrates a flow diagram for a method of FM demodulating an SAP signal in accordance with an embodiment of the present invention. First, at step 310, a BTSC composite signal 110 is transmitted to the BPF 120. Then, at step 320, the bandpass filter removes the stereo components, such as L+R, L-R, L, R, or professional channel, for example, to isolate the secondary audio program signal.

[0029] Next, at step 330, the SAP signal is sent to both the delay module 130 and the Hilbert filter 140. At step 340, the Hilbert filter is applied to the SAP signal to produce a phase quadrature signal, with a 90 degree difference in phase from the input signal. The output signal from the Hilbert filter 140 is signal Xq, represented by the following equation:

$$(1) \qquad Xq(t) = \sin[2\pi f_c t + 2\pi f_c M(t)],$$

wherein f_c represents the FM carrier frequency, t represents time, and M(t) represents the desired signal information.

[0030] Also, at step 350, the SAP signal is delayed by the delay module 130. The delay produces signal Xi, represented by the following equation:

$$(2) \qquad Xi(t) = \cos[2\pi f_c t + {}^{t}\ m(t)dt] = \cos[2\pi f_c t + 2\pi f_c M(t)].$$

Thus, Xq and Xi are separated by 90 degrees.

[0031] Then, at step 360, the delayed and phase quadrature signals are transmitted to the FM demodulator 150. If the FM carrier amplitude is not a constant, the FM carrier amplitude may be adjusted using automatic gain correction to normalize the amplitude and adjusted based on a comparison with a reference value. Then, at step 370, the SAP signal is demodulated using the delayed signal Xi and the quadrature signal Xq. The FM demodulator 150 uses the simplified demodulation equation:

(3) $m(n) \quad I(n)*Q(n-d) - Q(n)*I(n-d)$,

wherein I(n) represents the delayed signal Xi, Q(n) represents the quadrature phase signal Xq, d represents the non-unity delay, and n represents a discrete time index. The simplified demodulation equation is obtained from the more detailed demodulation equation based on Xq and Xi. The original equation is:

(4) $m_d(t) = [\{x_i(t)\}'x_q(t) - x_i(t)\{x_q(t)\}']/[\{x_i(t)\}^2 + \{x_q(t)\}^2]$.

Thus, the simplified FM demodulation approximation of Xi and Xq is:

(5) $m(n) \quad (x_i(n)x_q(n-d) - x_q(n)x_i(n-d))$.

Further detail on the use of the general delay d and the approximation is provided below.

[0032] Next, at step 380, the demodulated SAP signal is transmitted to the LPF 160. Finally, at step 390, the LPF 160 filters the demodulated signal to remove noise outside the range of the SAP signal. The resulting signal is the FM demodulated signal 180. The FM demodulated signal 180 is in digital form and may be used in televisions, radios, or other such devices.

[0033] Common digital approximations utilize a delay of one. Certain embodiments of the present invention employ a non-unity delay of greater than one. A general delay d, greater than one, may allow simplified FM demodulation of a variety of digital signals, such as SAP signals, Bluetooth™ communication signals, and other digital audio signals. The following equations illustrate the use of a general delay d. In the equations, m(n) represents an original message sequence, such as speech, music, program material, or data, for example. M(n) represents the original message sequence integrated modulo 2*pi. Also, $\tilde{m}(n)$ represents a scaled approximate message signal received using an approximate and efficient digital demodulation equation. I(n) and Q(n) are the in-phase and quadrature parts of the received FM modulated signal. In the equations, d is a delay used in the approximate demodulation equation (a non-negative integer), n is a discrete

time index, f_c is a carrier frequency, F_s is a sampling frequency, and f_{dev} is a frequency deviation of the FM modulation.

$$I(n) = \cos\left(2\pi \frac{f_c}{F_s} n + M(n)\right)$$

$$Q(n) = \sin\left(2\pi \frac{f_c}{F_s} n + M(n)\right)$$

$$M(n) = M(n-d) + 2\pi f_{dev} * m(n)$$

$$m(n) = \frac{M(n) - M(n-d)}{2\pi f_{dev} * d}$$

$$\widetilde{m}(n) = I(n) * Q(n-d) - I(n-d) * Q(n)$$

$$= \cos\left(2\pi \frac{f_c}{F_s} n + M(n)\right) \sin\left(2\pi \frac{f_c}{F_s}(n-d) + M(n-d)\right) - \sin\left(2\pi \frac{f_c}{F_s} n + M(n)\right) \cos\left(2\pi \frac{f_c}{F_s}(n-d) + M(n-d)\right)$$

$$= -\sin\left(2\pi \frac{f_c}{F_s}(d) + M(n) - M(n-d)\right)$$

$$= -\sin\big(M(n) - M(n-d)\big) \cos\left(2\pi \frac{f_c}{F_s} d\right) + \cos((M(n) - M(n-d)) \sin\left(2\pi \frac{f_c}{F_s} d\right)$$

$$\textit{Example}: (d = 2, Fs = 4 * f_c)$$

$$= \sin(M(n) - M(n-2)) \approx M(n) - M(n-2) = 4\pi f_{dev} * m(n)$$

[0034] Further details regarding the use of SAP and stereo signals or an example of how FM demodulation of an SAP channel fits in with a BTSC decoder may be found in the application entitled “System and Method of Performing Sample Rate Conversion of a Multi-Channel Audio Signal” filed under docket number 13587US01 on the same day as the application herein (docket number 13586US01) was filed, in the application entitled “System and Method of Performing Analog Multi-Channel Audio Signal Amplitude Correction” filed under docket number 13588US01 on the same day as the application herein (docket number 13586US01) was filed, in the application “Pilot Tone Based Automatic Gain Control System and Method” filed under docket number 13589US01 on the same day as the application herein (docket number 13586US01) was filed, and in the application “System and Method of Performing Digital Multi-Channel Audio Signal Decoding” filed under docket number 13578US01 on the same day as the application herein (docket number 13586US01) was filed.

[0035] In summary, certain embodiments of the present invention use minimal hardware and minimal equations to isolated and demodulate secondary audio program data included in a BTSC composite audio signal. The use of a simplified equation reduces the complexity of the process and also reduces the amount of hardware and memory involved in demodulation. Using a minimal number of filter taps also may reduce hardware complexity. Certain embodiments may improve the speed of the demodulation, as well as reduce the size, complexity, and cost of hardware. As a result, certain embodiments of the present invention afford an approach to achieve efficient, low cost, low power, digital audio signal decoding of frequency modulated information, such as SAP information, from digital signals, such as BTSC audio signals, in the digital domain.

[0036] While the invention has been described with reference to a preferred embodiment, it will be understood by those skilled in the art that various changes may be made and equivalents may be substituted without departing from the scope of the invention. In addition, many modifications may be made to adapt a particular situation or material to the teachings of the invention without departing from its scope. Therefore, it is intended that the invention not be limited to the particular embodiment disclosed, but that the invention will include all embodiments falling within the scope of the appended claims.